

ECOLE NATIONALE POLYTECHNIQUE

DEPARTEMENT DE L'ELECTRONIQUE :

Seminar Report:

Digital TV Transmission principles

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INTRODUCTION

-Why digital?

The main advantages of these digital formats are that they allow multiple copies to be made without any degradation in quality, and the creation of special effects not otherwise possible in analog format, and they simplify editing of all kinds, as well as permitting international exchange independent of the broadcast standard to be used for diffusion (NTSC, PAL, SECAM, D2-MAC, MPEG).

-More Channels:

More channels means larger bandwidth the disadvantage of greater bandwidth demands has been overcome by enormous advances in data compression techniques it's now possible to put many digital channels in the bandwidth occupied by one analogue channel Data compression and source coding techniques Allows larger bandwidth Bandwidth of one analog channel = bandwidth of 5 digital channels.

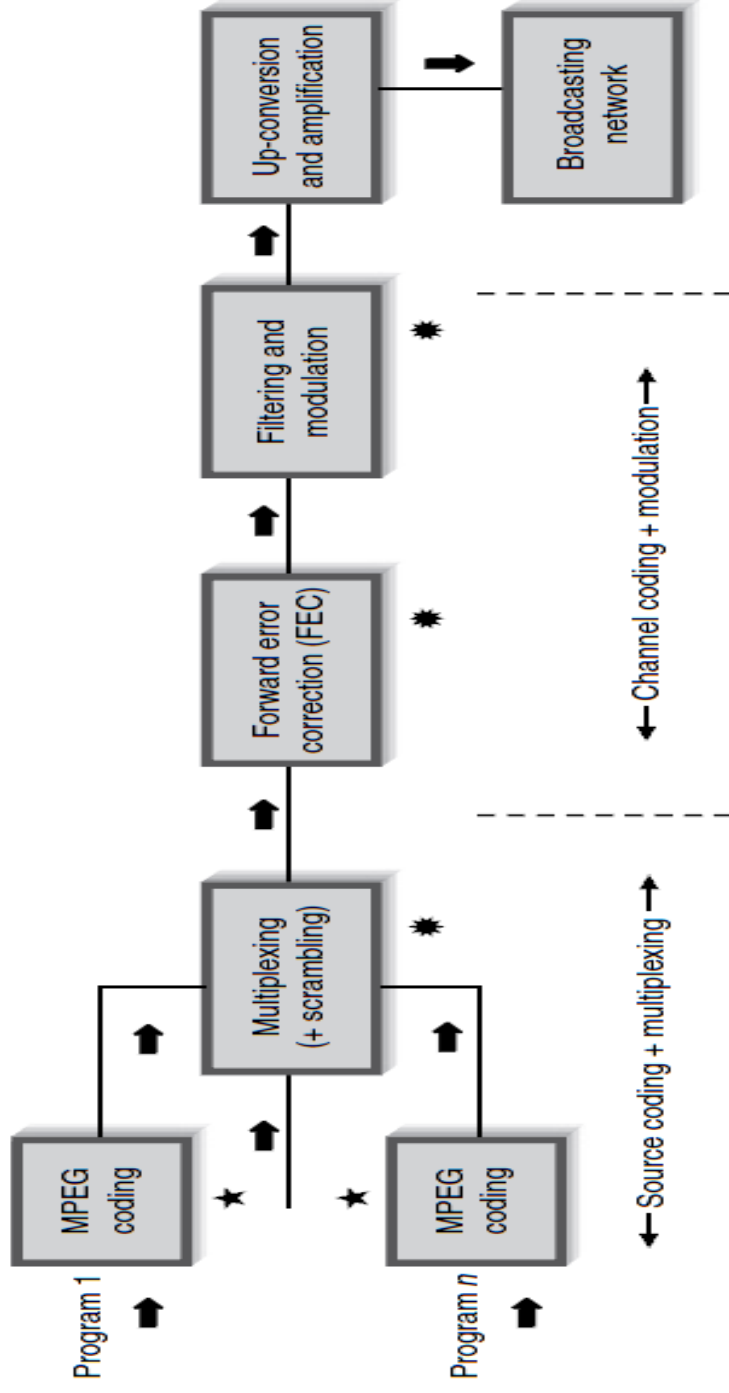


Fig1: Architecture of Digital transmission

2-SOURCE CODING

2-1. Some general data compression principles:

- Run length coding (RLC)

When an information source emits successive message elements which can deliver relatively long series of identical elements it is advantageous to transmit the code of this element and the number of successive occurrences rather than to repeat the code of the element.

- Variable length coding (VLC) or entropy coding

The method is based on the fact that the probability of occurrence of an element generated by a source and coded on n bits is sometimes not the same (i.e., equiprobable) for all elements among the 2^n different possibilities. This means that, in order to reduce the bit-rate required to transmit the sequences generated by the source, it is advantageous to encode the most frequent elements with less than n bits and the less frequent elements with more bits, resulting in an average length that is less than a fixed length of n bits.

2-2. Compression of moving pictures (MPEG)

In 1990, the need to store and reproduce moving pictures and the associated sound in digital format pushed ISO to form an expert group called (Motion Pictures Experts Group). In addition to the intrinsic spatial redundancy exploited by JPEG for fixed pictures (Lenouar), the coding of moving pictures allows exploitation of the very important temporal redundancy between successive pictures which make up a video sequence.

2-2-1.MPEG-1 :medium quality video bit-rate of 1.5 Mb/s for

The different types of MPEG pictures:

1-I (intra) pictures

-Same as jpeg

-Has low compression rate.

2-P (predicted) pictures

-From I or P using “motion compensation “

-higher compression rate.

3-B bi-directional pictures.

-Interpolation between two I or P pictures

the main objective for MPEG-1 was to reach a medium quality video with a constant total bit-rate of 1.5 Mb/s for storing video and audio on CD-ROM.they contain all the information necessary for their reconstruction by the Decoder.

P (predicted) pictures are coded from the preceding I or P picture,using the techniques of motion-compensated prediction coded by bi-directional interpolation between the I or P picture which precedes and follows them.

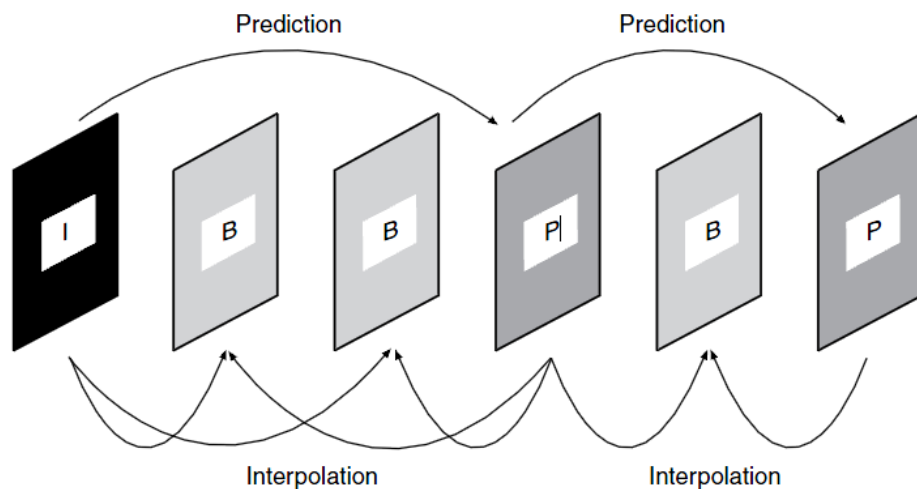
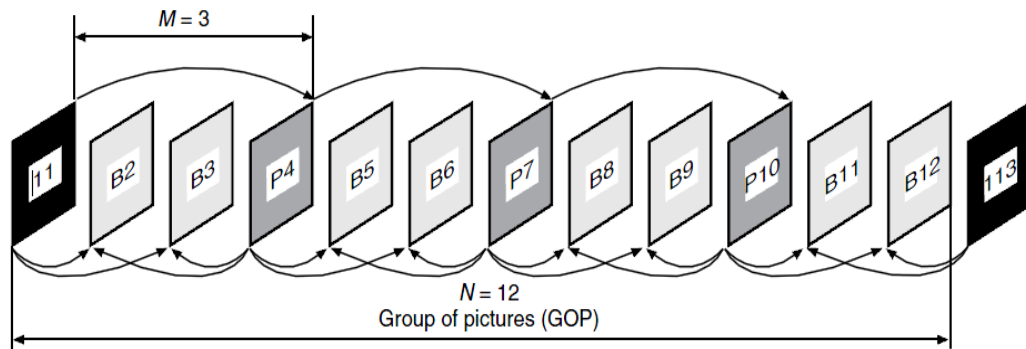


Fig:2-a



Example of an MPEG group of pictures for $M = 3$ and $N = 12$.

Fig:2-b

There are two parameters M and N to describe the succession

M between two I or P pictures

N between two I pictures

2-2-2. Motion compensation:

Basically there is no correlation between two sequenced pictures. Motion compensation defines a *motion vector* which ensures the correlation between an arrival zone on the second picture and a departure zone on the first picture using block matching techniques.

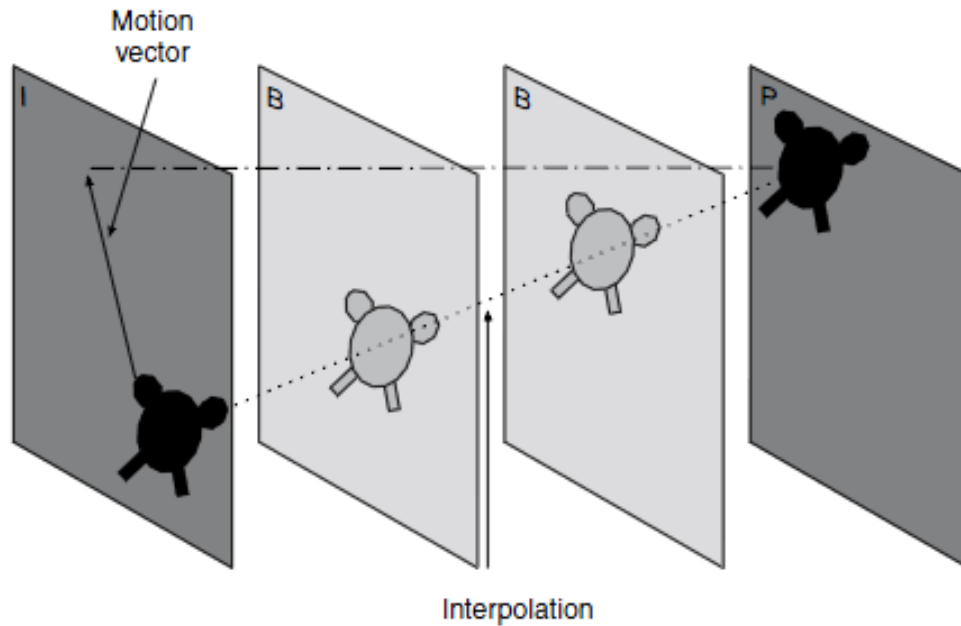


Fig:3. Interpolation

Compression of audio signals

- ❑ Notion of frequency masking and temporal masking

In the case of two signals of relatively near frequencies, The difference between strength of them lead the ear to hear just one of them .

- ❑ psychoacoustic model quantify this aspects

A sound of strong amplitude also masks sounds immediately preceding it or following it in time.

- ❑ The MPEG audio layers:

The MPEG audio standard defines three coding layers which offer very different compression rates for a given perceived audio quality.

-layer 1:MUSICAM

-layer 2

-layer 3:mp3

Video coding with MPEG-2

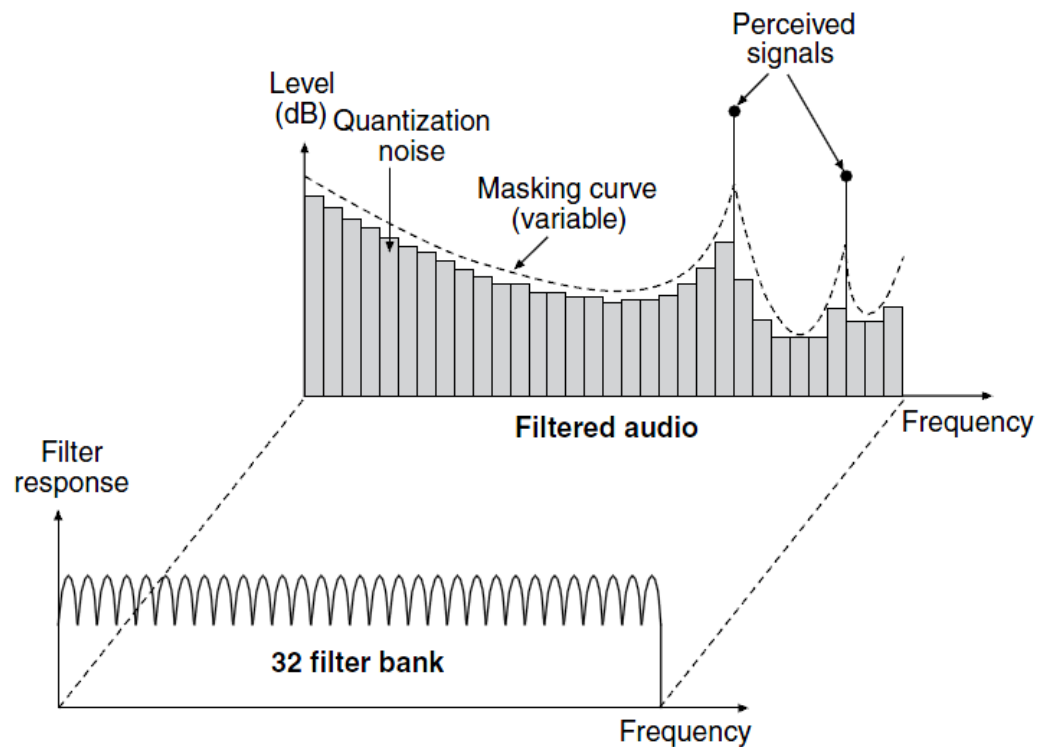


Fig:4. The :MPEG-2 Spectrum

The principle of the coding process consists of first dividing the audio frequency band into 32 sub-bands of equal width by means of a *polyphase* filter bank. The output signal from a sub-band filter corresponding to a duration of 32 **PCM** samples is called a **sub-band sample** as illustrated.

3-SOURCE MULTIPLEXING:

Organization of the MPEG-1 multiplex: system layer

- ❑ The transport stream: based on the use of fixed-length (188 byte) transport stream

-packets identified by headers

- ❑ Each header identifies a particular application

bitstream (also called an elementary bitstream, or ES) which represents real, coded, ‘entertainment’ information.

- ❑ The elementary bitstreams for video and audio are themselves wrapped in a variablelength packet structure called the packetized elementary stream (PES)

Audio and video encoders deliver as their output elementary streams(ES) which are the constituents of the so-called compression layer.

Each elementary stream carries access units (AU) which are the coded representations of presentation units (PU) These bitstreams, as well as other streams carrying other *private data*, have to be combined in an organized manner and supplemented with additional information to allow their separation by the decoder, synchronization of picture and sound, and selection by the user of the particular components of interest.

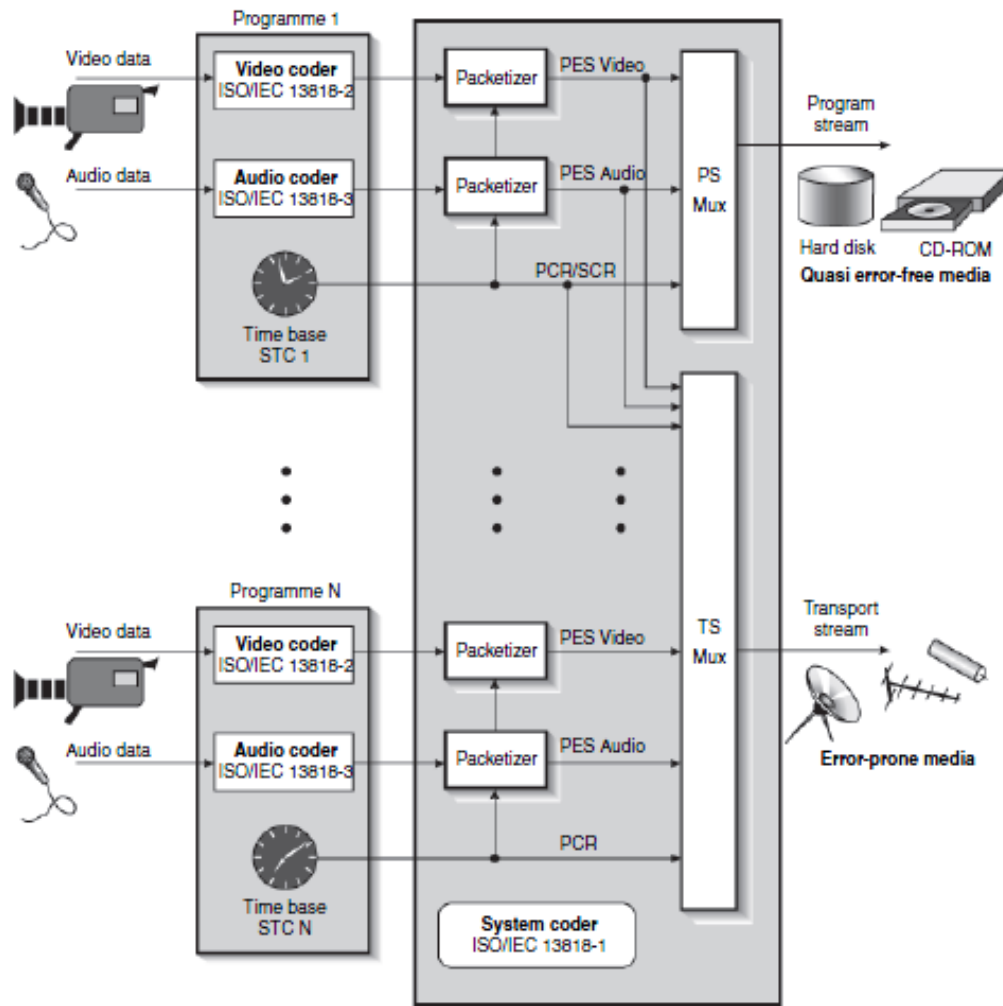


Fig :5. Source Multiplexing

3-2.Scrambling and conditional access:

Why conditional access?

The proportion of free access programs among analog TV transmissions by cable or satellite is decreasing continuously, at the same time as their number increases; hence, it is almost certain that the majority of digital TV programs will be pay-TV services, in order to recover as quickly as possible the high investments required to launch these services.

The scrambling algorithm envisaged to resist attacks from hackers for as long as possible consists of a cipher with two layers, each palliating the weaknesses of the other:

- a *block layer* using blocks of 8 bytes (reverse cipher block chaining mode),
- a *stream layer* (pseudo-random byte generator).

4-MODULATION BY DIGITAL SIGNALS

Digital tv modulation depends on transmission technique

❑ Cable (DVB-C): 16 to 256 QAM

Channel width 6, 7, or 8MHz

❑ Satellite (DVB-S): QPSK

Channel width 27 ~ 36MHz

❑ Sat. (DVB-S2): QPSK to 32 APSK

Channel width 27 ~ 36MHz

❑ Terr. (DVB-T): OFDM 2K/8K

Channel width 6, 7, or 8MHz

❑ Mobile (DVB-H): OFDM 2K/4K/8K

Channel width 6, 7, or 8MHz

CONCLUSION:

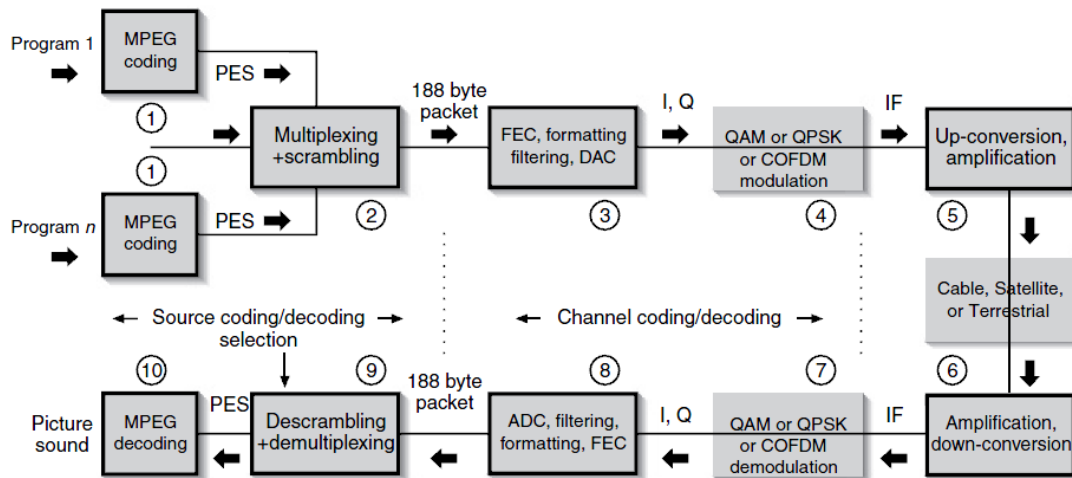


Fig: A simplified view of the complete DVB transmission/reception chain

1. Video and audio signals of the programs to be broadcast are each put through an MPEG-2 encoder which delivers the video and audio PESs to the multiplexer (about four to eight programs per RF channel depending on the parameters chosen for the encoding).
2. These PESs are used by the multiplexer to form 188 byte transport packets, which are eventually scrambled (CAT tables carrying conditional access information ECM/EMM are inserted in this case), as well as the PAT, PMT, PSI, and DVB-SI tables for the electronic program guide (EPG).
3. The RS error correction increases the packet length to 204 bytes; in the case of satellite, the convolutional coding further multiplies the bit-rate by a factor of between 1.14 ($R_c = 7/8$) and 2 ($R_c = 1/2$); formatting of the data (*symbol mapping*) followed by filtering and D/A conversion produce the I and Q analog signals.
4. The I and Q signals modulate in QPSK (satellite) and QAM (cable) or COFDM (terrestrial) an IF carrier (intermediate frequency of the order of 70 MHz). 5. This IF is up-converted into the appropriate frequency band (depending on the medium) for transmission to the end users.